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(54) **ADAPTIVE GAIN CONTROL FOR DIGITAL AUDIO SAMPLES IN A MEDIA STREAM**

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CPC ***H04R 3/00*** (2013.01); ***G10L 21/0364*** (2013.01); ***H03G 7/007*** (2013.01); *H04S 2400/13* (2013.01)

(58) **Field of Classification Search**

CPC G10L 21/0364; H03G 7/007; H04R 3/00; H04S 2400/13

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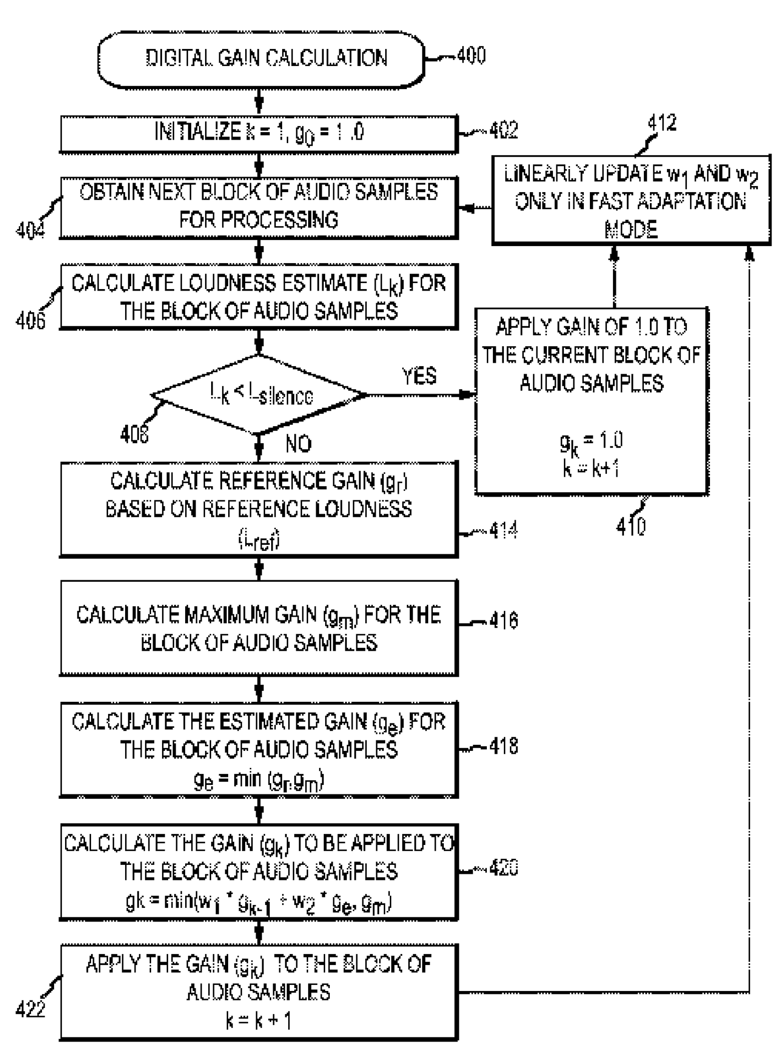
(57) **ABSTRACT**

An adaptive gain control system and related operating method for digital audio samples is provided. The method is suitable for use with a digital media encoding system that transmits encoded media streams to a remotely-located presentation device such as a media player. The method begins by initializing the processing of a media stream. Then, the method adjusts the gain of a first set of digital audio samples in the media stream using a fast gain adaptation scheme, resulting in a first group of gain-adjusted digital audio samples having normalized volume during presentation. The method continues by adjusting the gain of a second set of digital audio samples in the media stream using a steady state gain adaptation scheme that is different than the fast gain adaptation scheme, resulting in a second group of gain-adjusted digital audio samples having normalized volume during presentation.

18 Claims, 4 Drawing Sheets



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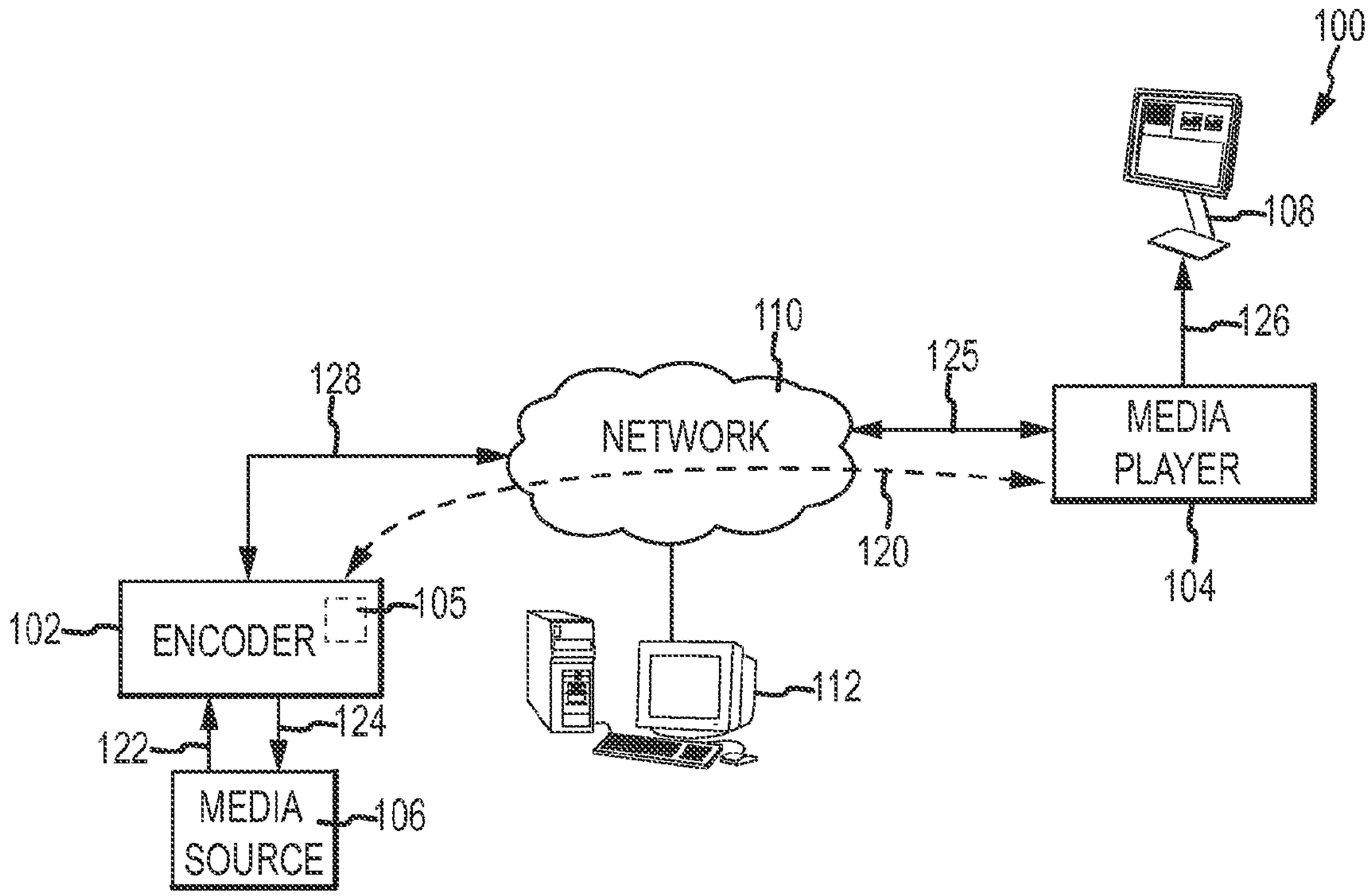
100
108
126
110
125
128
NETWORK
MEDIA
PLAYER
120
104
102
ENCODER
105
112
122
124
MEDIA
SOURCE
106

FIG.1

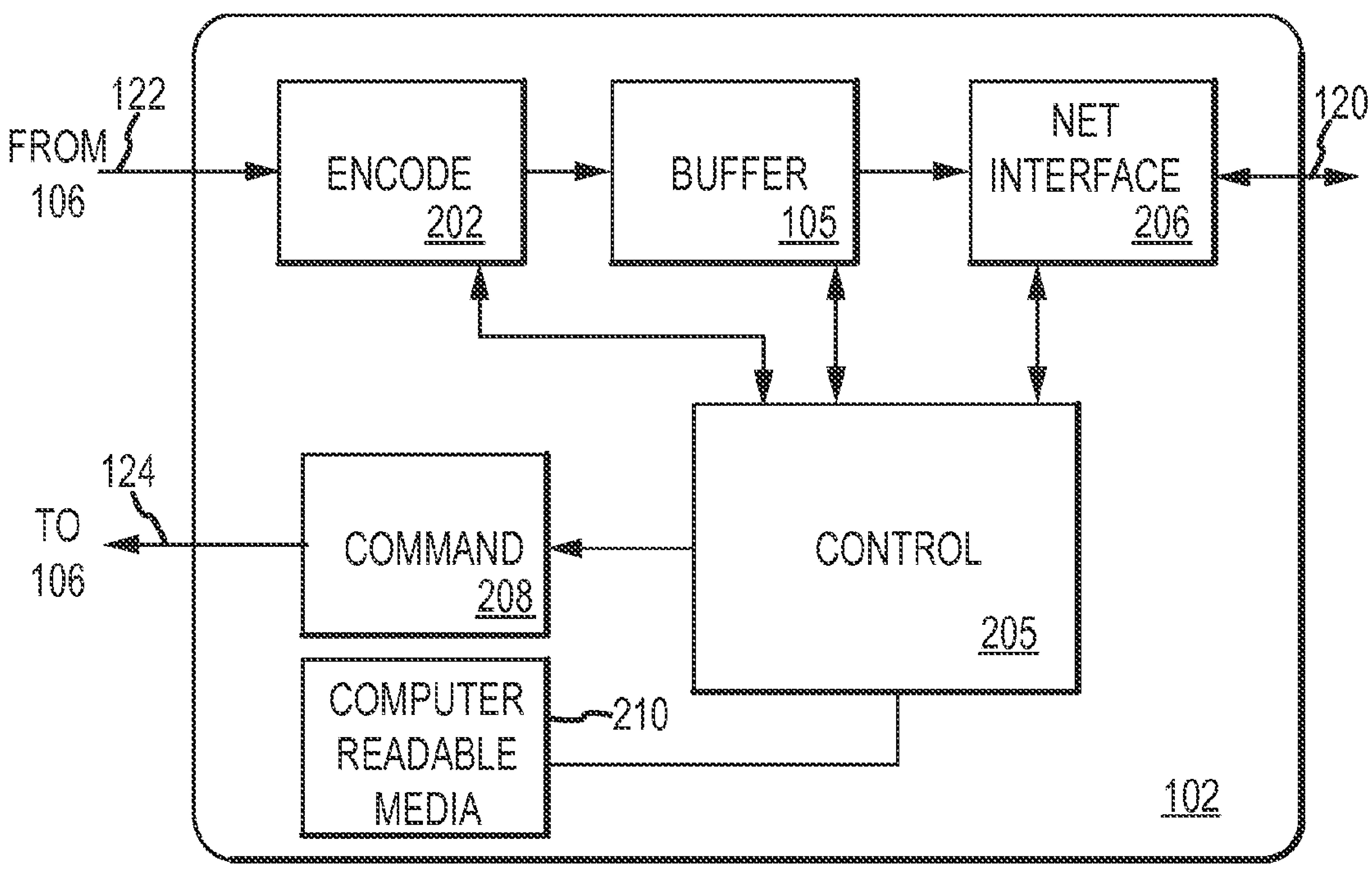
122
FROM
106
ENCODE
202
BUFFER
105
NET
INTERFACE
206
120
124
TO
106
COMMAND
208
CONTROL
205
COMPUTER
READABLE
MEDIA
210
102

FIG.2

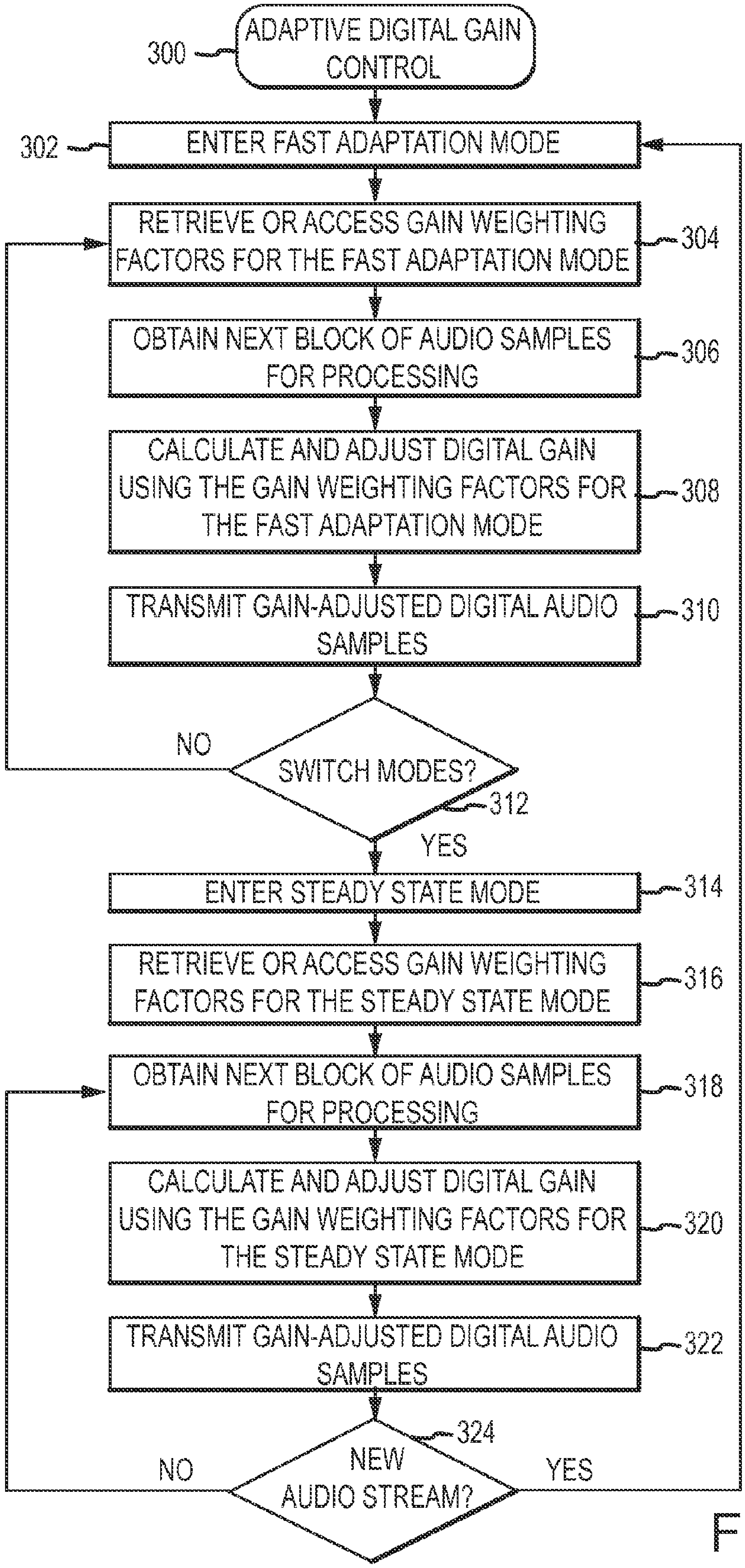
ADAPTIVE DIGITAL GAIN CONTROL
300
ENTER FAST ADAPTATION MODE
302
RETRIEVE OR ACCESS GAIN WEIGHTING FACTORS FOR THE FAST ADAPTATION MODE
304
OBTAIN NEXT BLOCK OF AUDIO SAMPLES FOR PROCESSING
306
CALCULATE AND ADJUST DIGITAL GAIN USING THE GAIN WEIGHTING FACTORS FOR THE FAST ADAPTATION MODE
308
TRANSMIT GAIN-ADJUSTED DIGITAL AUDIO SAMPLES
310
NO
SWITCH MODES?
312
YES
ENTER STEADY STATE MODE
314
RETRIEVE OR ACCESS GAIN WEIGHTING FACTORS FOR THE STEADY STATE MODE
316
OBTAIN NEXT BLOCK OF AUDIO SAMPLES FOR PROCESSING
318
CALCULATE AND ADJUST DIGITAL GAIN USING THE GAIN WEIGHTING FACTORS FOR THE STEADY STATE MODE
320
TRANSMIT GAIN-ADJUSTED DIGITAL AUDIO SAMPLES
322
324
NO
NEW AUDIO STREAM?
YES

FIG.3

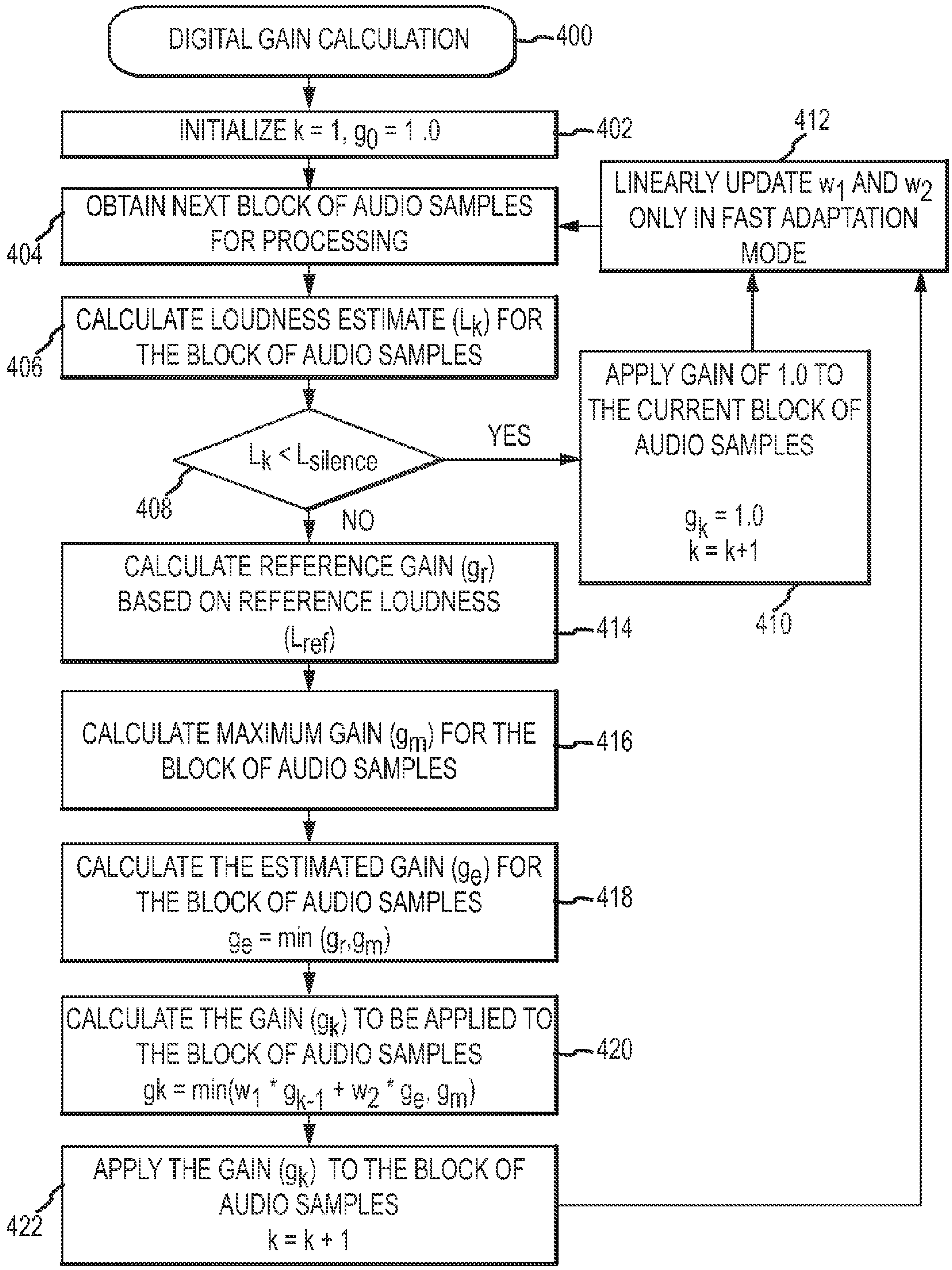
DIGITAL GAIN CALCULATION
400
INITIALIZE k = 1, g_0 = 1 .0
402
OBTAIN NEXT BLOCK OF AUDIO SAMPLES FOR PROCESSING
404
CALCULATE LOUDNESS ESTIMATE (L_k) FOR THE BLOCK OF AUDIO SAMPLES
406
$L_k < L_{silence}$
408
YES
NO
APPLY GAIN OF 1.0 TO THE CURRENT BLOCK OF AUDIO SAMPLES
$g_k = 1.0$
$k = k+1$
410
LINEARLY UPDATE w_1 AND w_2 ONLY IN FAST ADAPTATION MODE
412
CALCULATE REFERENCE GAIN (g_r) BASED ON REFERENCE LOUDNESS (L_{ref})
414
CALCULATE MAXIMUM GAIN (g_m) FOR THE BLOCK OF AUDIO SAMPLES
416
CALCULATE THE ESTIMATED GAIN (g_e) FOR THE BLOCK OF AUDIO SAMPLES
$g_e = \min(g_r, g_m)$
418
CALCULATE THE GAIN (g_k) TO BE APPLIED TO THE BLOCK OF AUDIO SAMPLES
$gk = \min(w_1 * g_{k-1} + w_2 * g_e, g_m)$
420
APPLY THE GAIN (g_k) TO THE BLOCK OF AUDIO SAMPLES
$k = k + 1$
422

FIG.4

ADAPTIVE GAIN CONTROL FOR DIGITAL AUDIO SAMPLES IN A MEDIA STREAM

CROSS-REFERENCE TO RELATED APPLICATION

This application is a divisional of U.S. patent application Ser. No. 12/507,971, filed Jul. 23, 2009, and issued on Mar. 26, 2013 as U.S. Pat. No. 8,406,431.

TECHNICAL FIELD

Embodiments of the subject matter described herein relate generally to the processing of digital audio samples in a media stream. More particularly, embodiments of the subject matter relate to digitally adjusting gain of digital audio samples in a media stream such that the perceived volume is normalized during presentation.

BACKGROUND

Recently, consumers have expressed significant interest in "place-shifting" devices that allow viewing of television or other media content at locations other than their primary media presentation device. Place-shifting devices typically packetize media content that can be transmitted over a local or wide area network to a portable computer, mobile phone, personal digital assistant, remote television or other remote device capable of playing back the packetized media stream for the viewer. Place-shifting therefore allows consumers to view their media content from remote locations such as other rooms, hotels, offices, and/or any other locations where portable media player devices can gain access to a wireless or other communications network.

While place-shifting does greatly improve the convenience afforded to the end user, there remain some challenges related to the manner in which different media streams are presented at the end device. For instance, the digital audio samples in one media stream may be associated with a baseline or average presentation loudness or volume, while the digital audio samples in another media stream may be associated with a different baseline/average presentation loudness or volume. Thus, if the user switches between different media streams the perceived loudness may be inconsistent, and the user will therefore need to adjust the volume control on the presentation device.

Volume normalization techniques can be utilized to automatically adjust the volume perceived by the user. Some volume normalization techniques operate in the analog domain, and others operate in the digital domain. Digital volume normalization techniques are best suited for place-shifting applications because the media streams are encoded and transmitted to the presentation device using data packets. Unfortunately, existing digital volume normalization techniques tend to be ineffective and/or they introduce audible artifacts that can be distracting to the user.

BRIEF SUMMARY

An adaptive gain control method for digital audio samples is provided. The method begins by initializing processing of a media stream. The method continues by adjusting gain of a first set of digital audio samples in the media stream using a fast gain adaptation scheme, resulting in a first group of gain-adjusted digital audio samples having normalized volume during presentation. Thereafter, the method adjusts gain of a second set of digital audio samples in the media stream using a steady state gain adaptation scheme that is different than the fast gain adaptation scheme, resulting in a second group of gain-adjusted digital audio samples having normalized volume during presentation.

Also provided is a computer program product, which is tangibly embodied in a computer-readable medium. The computer program product is operable to cause a digital media processing device to perform operations for a media stream. These operations include: calculating a loudness estimate for a current block of digital audio samples in the media stream; calculating a reference gain value for the current block of digital audio samples, the reference gain value being influenced by the loudness estimate; calculating a maximum gain value for the current block of digital audio samples; calculating an estimated gain value for the current block of digital audio samples, the estimated gain value being influenced by the reference gain value and the maximum gain value; and calculating a gain value for the current block of digital audio samples, the gain value being influenced by the estimated gain value, the maximum gain value, and a previous gain value for a previous block of digital audio samples in the media stream. The computer program product is also operable to cause the digital media processing device to modify the current block of digital audio samples by applying the gain value to the digital audio samples in the current block of digital audio samples. In certain embodiments the maximum gain value is influenced by dynamic range of the current block of digital audio samples.

A system for processing digital audio samples in a media stream is also provided. The system includes a first means for adjusting gain of a first block of digital audio samples in the media stream using a fast gain adaptation scheme, resulting in a first block of gain-adjusted digital audio samples. The system also includes a second means for adjusting gain of a second block of digital audio samples in the media stream using a steady state gain adaptation scheme that is different than the fast gain adaptation scheme, resulting in a second block of gain-adjusted digital audio samples. The system also includes means for transmitting gain-adjusted digital audio samples to a remotely-located media player.

This summary is provided to introduce a selection of concepts in a simplified form that are further described below in the detailed description. This summary is not intended to identify key features or essential features of the claimed subject matter, nor is it intended to be used as an aid in determining the scope of the claimed subject matter.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete understanding of the subject matter may be derived by referring to the detailed description and claims when considered in conjunction with the following figures, wherein like reference numbers refer to similar elements throughout the figures.

FIG. **1** is a schematic representation of an embodiment of a media presentation system;

FIG. **2** is a schematic representation of an embodiment of a digital media processing device;

FIG. **3** is a flow chart that illustrates an embodiment of an adaptive digital gain control process; and

FIG. **4** is a flow chart that illustrates an embodiment of a digital gain calculation process.

DETAILED DESCRIPTION

The following detailed description is merely illustrative in nature and is not intended to limit the embodiments of the subject matter or the application and uses of such embodiments. As used herein, the word "exemplary" means "serving as an example, instance, or illustration." Any implementation described herein as exemplary is not necessarily to be construed as preferred or advantageous over other implementations. Furthermore, there is no intention to be bound by any expressed or implied theory presented in the preceding technical field, background, brief summary or the following detailed description.

Techniques and technologies may be described herein in terms of functional and/or logical block components, and with reference to symbolic representations of operations, processing tasks, and functions that may be performed by various computing components or devices. Such operations, tasks, and functions are sometimes referred to as being computer-executed, computerized, software-implemented, or computer-implemented. In practice, one or more processor devices can carry out the described operations, tasks, and functions by manipulating electrical signals representing data bits at memory locations in the system memory, as well as other processing of signals. The memory locations where data bits are maintained are physical locations that have particular electrical, magnetic, optical, or organic properties corresponding to the data bits. It should be appreciated that the various block components shown in the figures may be realized by any number of hardware, software, and/or firmware components configured to perform the specified functions. For example, an embodiment of a system or a component may employ various integrated circuit components, e.g., memory elements, digital signal processing elements, logic elements, look-up tables, or the like, which may carry out a variety of functions under the control of one or more microprocessors or other control devices.

When implemented in software or firmware, various elements of the systems described herein are essentially the code segments or instructions that perform the various tasks. The program or code segments can be stored in a processor-readable medium or transmitted by a computer data signal embodied in a carrier wave over a transmission medium or communication path. The "processor-readable medium" or "machine-readable medium" may include any medium that can store or transfer information. Examples of the processor-readable medium include an electronic circuit, a semiconductor memory device, a ROM, a flash memory, an erasable ROM (EROM), a floppy diskette, a CD-ROM, an optical disk, a hard disk, or the like. The computer data signal may include any signal that can propagate over a transmission medium such as electronic network channels, optical fibers, air, electromagnetic paths, or RF links. The code segments may be downloaded via computer networks such as the Internet, an intranet, a LAN, or the like.

According to various embodiments, the perceived presentation loudness (i.e., volume) of a media stream is normalized or leveled relative to a reference loudness, such that different media streams are presented at about the same average loudness for a constant volume setting at the presentation device. The volume normalization scheme is carried out in the digital domain by modifying, adjusting, or otherwise altering the digital audio samples associated with the media streams. In certain embodiments, the digital audio samples are modified by a digital media processing device that encodes and transmits media streams (via a data communication network) to the user's media presentation device (e.g., a laptop computer, a cell phone, a remote set-top box, or the like). The digitally normalized audio samples are transmitted to the presentation device in the desired media stream, resulting in normalized presentation volume for different media streams. Notably, the presentation device itself need not be modified to support the digital volume normalization techniques described here because the digital audio samples arrive at the presentation device after application of digital gain adjustment.

Turning now to the figures and with initial reference to FIG. **1**, an exemplary embodiment of a media presentation system **100** can be utilized to carry out place-shifting of digital media content that includes digital audio samples. This particular embodiment of the system **100** includes a digital media processing device (e.g., a place-shifting encoder system **102**) that receives media content **122** from a content source **106**, encodes the received content into a streaming format, and then transmits the encoded media stream **120** to a remotely-located digital media player (or other presentation device) **104** over a network **110**. The media player **104** receives the encoded media stream **120**, decodes the stream, and presents the decoded content to a viewer on a television or other display **108**. Although not depicted in FIG. **1**, the media player **104** includes or cooperates with at least one speaker, audio transducer, or other sound-generating element that supports the presentation of the audio portion of media streams. In various embodiments, a server **112** may also be provided to communicate with the encoder system **102** and/or the media player **104** via the network **110** to assist these devices in locating each other, maintaining security, providing or receiving content or information, and/or any other features as desired. This feature is not required in all embodiments, however, and the concepts described herein may be deployed in any data streaming application or environment, including place-shifting but also any other media or other data streaming situation.

The encoder system **102** is any component, hardware, software logic and/or the like capable of transmitting a packetized stream of media content over the network **110**. In various embodiments, the encoder system **102** incorporates suitable encoder and/or transcoder (collectively "encoder") logic to convert audio/video or other media content **122** into a packetized format that can be transmitted over the network **110**. The media content **122** may be received in any format, and may be received from any internal or external content source **106** such as any sort of broadcast, cable or satellite television programming source, a "video-on-demand" or similar source, a digital video disk (DVD) or other removable media, a video camera, and/or the like. The encoder system **102** encodes the media content **122** to create the encoded media stream **120** in any manner. In various embodiments, the encoder system **102** contains a transmit buffer **105** that temporarily stores encoded data prior to transmission on the network **110**.

In practice, an embodiment of the encoder system **102** may be implemented using any of the various SLINGBOX products available from Sling Media of Foster City, Calif., although other products could be used in other embodiments. Certain embodiments of the encoder system **102** are generally capable of receiving the media content **122** from an external content source **106** such as any sort of digital video recorder (DVR), set top box (STB), cable or satellite programming source, DVD player, and/or the like. In such embodiments, the encoder system **102** may additionally provide commands **124** to the content source **106** to produce the desired media content **122**. Such commands **124** may be provided over any sort of wired or wireless interface, such as an infrared or other wireless transmitter that emulates remote control commands receivable by the content source

106. Other embodiments, however, particularly those that do not involve place-shifting, may modify or omit this feature entirely.

In other embodiments, the encoder system **102** may be integrated with any sort of content-receiving or other capabilities typically affiliated with the content source **106**. The encoder system **102** may be a hybrid STB or other receiver, for example, that also provides transcoding and place-shifting features. Such a device may receive satellite, cable, broadcast and/or other signals that encode television programming or other content received from an antenna, modem, server and/or other source. A receiver of the encoder system **102** may further demodulate or otherwise decode the received signals to extract programming that can be locally viewed and/or place-shifted to the remotely-located media player **104** as appropriate. In this regard, the encoder system **102** may also include a content database stored on a hard disk drive, memory, or other storage medium to support a personal or digital video recorder (DVR) feature or other content library as appropriate. Hence, in some embodiments, the content source **106** and the encoder system **102** may be physically and/or logically contained within a common component, housing or chassis.

In still other embodiments, the encoder system **102** includes or is implemented as a software program, applet, or the like executing on a conventional computing system (e.g., a personal computer). In such embodiments, the encoder system **102** may encode, for example, some or all of a screen display typically provided to a user of the computing system for place-shifting to a remote location. One device capable of providing such functionality is the SlingProjector product available from Sling Media of Foster City, Calif., which executes on a conventional personal computer, although other products could be used as well.

The media player **104** is any device, component, module, hardware, software and/or the like capable of receiving the encoded media stream **120** from one or more encoder systems **102**. In various embodiments, the media player **104** is personal computer (e.g., a "laptop" or similarly portable computer, although desktop-type computers could also be used), a mobile phone, a personal digital assistant, a personal media player, or the like. In many embodiments, the media player **104** is a general purpose computing device that includes a media player application in software or firmware that is capable of securely connecting to the encoder system **102**, and is capable of receiving and presenting media content to the user of the device as appropriate. In other embodiments, however, the media player **104** is a standalone or other separate hardware device capable of receiving the encoded media stream **120** via any portion of the network **110** and decoding the encoded media stream **120** to provide an output signal **126** that is presented on the display **108**. One example of a standalone media player **104** is the SLINGCATCHER product available from Sling Media of Foster City, Calif., although other products could be equivalently used.

The network **110** is any digital or other communications network capable of transmitting messages between senders (e.g., the encoder system **102**) and receivers (e.g., the media player **104**). In various embodiments, the network **110** includes any number of public or private data connections, links or networks supporting any number of communications protocols. The network **110** may include the Internet, for example, or any other network based upon TCP/IP or other conventional protocols. In various embodiments, the network **110** also incorporates a wireless and/or wired telephone network, such as a cellular communications network for communicating with mobile phones, personal digital assistants, and/or the like. The network **110** may also incorporate any sort of wireless or wired local area networks, such as one or more IEEE 802.3 and/or IEEE 802.11 networks.

The encoder system **102** and/or the media player **104** are therefore able to communicate in any manner with the network **110** (e.g., using any sort of data connections **128** and/or **125**, respectively). Such communication may take place over a wide area link that includes the Internet and/or a telephone network, for example; in other embodiments, communications between the encoder system **102** and the media player **104** may take place over one or more wired or wireless local area links that are conceptually incorporated within the network **110**. In various equivalent embodiments, the encoder system **102** and the media player **104** may be directly connected via any sort of cable (e.g., an Ethernet cable or the like) with little or no other network functionality provided.

Many different place-shifting scenarios could be formulated based upon available computing and communications resources, consumer demand and/or any other factors. In various embodiments, consumers may wish to place-shift content within a home, office or other structure, such as from the encoder system **102** to a desktop or portable computer located in another room. In such embodiments, the content stream will typically be provided over a wired or wireless local area network operating within the structure. In other embodiments, consumers may wish to place-shift content over a broadband or similar network connection from a primary location to a computer or other remote media player **104** located in a second home, office, hotel or other remote location. In still other embodiments, consumers may wish to place-shift content to a mobile phone, personal digital assistant, media player, video game player, automotive or other vehicle media player, and/or other device via a mobile link (e.g., a GSM/EDGE or CDMA/EVDO connection, any sort of 3G or subsequent telephone link, an IEEE 802.11 "Wi-Fi" link, and/or the like). Several examples of place-shifting applications available for various platforms are provided by Sling Media of Foster City, Calif., although the concepts described herein could be used in conjunction with products and services available from any source.

FIG. **2** is a schematic representation of an embodiment of a digital media processing device, such as the encoder system **102**. Again, the encoder system **102** generally creates an encoded media stream **120** that is routable on the network **110** based upon the media content **122** received from the content source **106**. In this regard, and with reference now to FIG. **2**, the encoder system **102** typically includes an encoding module **202**, a transmit buffer **105**, and a network interface **206** in conjunction with appropriate control logic, which may be associated with a control module **205**. In operation, the encoding module **202** typically receives the media content **122** from the internal or external content source **106**, encodes the data into the desired format for the encoded media stream **120**, and stores the encoded data in the transmit buffer **105**. The network interface **206** then retrieves the formatted data from the transmit buffer **105** for transmission on the network **110**. The control module **205** suitably monitors and controls the encoding and network transmit processes carried out by the encoding module **202** and the network interface **206**, respectively, and may perform other functions as well. The encoder system **102** may also have a command module **208** or other feature capable of generating and providing the commands **124** to the content source **106**, as described above.

As noted above, creating the encoded media stream **120** typically involves encoding and/or transcoding the media content **122** received from the content source **106** into a suitable digital format that can be transmitted on the network **110**. Generally, the encoded media stream **120** is placed into a standard or other known format (e.g., the WINDOWS MEDIA format available from the Microsoft Corporation of Redmond, Wash., the QUICKTIME format, the REALPLAYER format, an MPEG format, and/or the like) that can be transmitted on the network **110**. This encoding may take place, for example, in any sort of encoding module **202** as appropriate. The encoding module **202** may be any sort of hardware (e.g., a digital signal processor or other integrated circuit used for media encoding), software (e.g., software or firmware programming used for media encoding that executes at the encoder system **102**), or the like. The encoding module **202** is therefore any feature that receives the media content **122** from content source **106** (e.g., via any sort of hardware and/or software interface) and encodes or transcodes the received data into the desired format for transmission on the network **110**. Although FIG. **2** shows a single encoding module **202**, in practice the encoder system **102** may include any number of encoding modules **202**. Different encoding modules **202** may be selected based upon the type of media player **104**, network conditions, user preferences, and/or the like.

In various embodiments, the encoding module **202** may also apply other modifications, transforms, and/or filters to the received content before or during the encoding/transcoding process. Video signals, for example, may be resized, cropped and/or skewed. Similarly, the color, hue and/or saturation of the signal may be altered, and/or noise reduction or other filtering may be applied. Digital rights management encoding and/or decoding may also be applied in some embodiments, and/or other features may be applied as desired. Audio signals may be modified by adjusting sampling rate, mono/stereo parameters, noise reduction, multichannel sound parameters and/or the like. In this regard, digital audio samples in a media stream can be modified in accordance with the adaptive digital gain control techniques and methodologies described in more detail below. Such gain control techniques can be used to modify blocks of digital audio samples to normalize the volume or loudness perceived by the user during presentation of the media stream.

The network interface **206** refers to any hardware, software and/or firmware that allows the encoder system **102** to communicate on the network **110**. In various embodiments, the network interface **206** includes suitable network stack programming and other features and/or conventional network interface (NIC) hardware such as any wired or wireless interface as desired.

In various embodiments, the control module **205** monitors and controls the encoding and transmit processes performed by the encoding module **202** and the network interface **206**, respectively. To that end, the control module **205** is any hardware, software, firmware or combination thereof capable of performing such features. In various embodiments, the control module **205** further processes commands received from the remote media player via the network interface **206** (e.g., by sending the commands **124** to the content source **106** via the command module **208** or the like). The control module **205** may also transmit commands to the media player **104** via the network interface **206** and/or may control or otherwise effect any other operations of the encoder system **102**. In various embodiments, the control module **205** implements the control features used to monitor and adjust the operation of the encoding module **202** and/or the network interface **206** to efficiently provide the media stream to the media player **104**.

Certain embodiments of the encoding module **202** may include or execute one or more computer programs (e.g., software) that are tangibly embodied in appropriately configured computer-readable media **210**. The computer program product is operable to cause the encoder system **102** to perform certain operations on media streams, as described in more detail below with reference to FIG. **3** and FIG. **4**. For this embodiment, FIG. **2** depicts the computer-readable media **210** associated with the control module **205**, although such an association need not be employed in all implementations. Indeed, the computer-readable media **210** may alternatively (or additionally) be utilized in connection with the encoding module **202** if so desired. As mentioned above, the computer-readable media **210** may include, without limitation: an electronic circuit, a semiconductor memory device, a ROM, a flash memory, an erasable ROM (EROM), a floppy diskette, a CD-ROM, an optical disk, a hard disk, or the like.

FIG. **3** is a flow chart that illustrates an embodiment of an adaptive digital gain control process **300**, and FIG. **4** is a flow chart that illustrates an embodiment of a digital gain calculation process **400**. The various tasks performed in connection with an illustrated process may be performed by software, hardware, firmware, or any combination thereof. The operations and instructions associated with a described process may be executed by any processor and/or other processing features within the encoder system **102**, and the particular means used to implement each of the various functions shown in the figures, then, could be any sort of processing hardware (such as the control module **205** of FIG. **2**) executing software or processor-based logic in any format. It should be appreciated that a described process may include any number of additional or alternative tasks, or may omit one or more illustrated tasks. Moreover, the tasks shown in the figures need not be performed in the illustrated order, and a described process may be incorporated into a more comprehensive procedure or process having additional functionality not described in detail herein.

Referring now to FIG. **3**, the adaptive digital gain control process **300** can be started when a new media stream has been designated for encoding and delivery to a presentation device. A media stream may include or otherwise be associated with a plurality of digital audio samples. As used here, a digital audio sample corresponds to a digital representation of an analog audio signal taken at a point in time (or over a relatively short period of time), as is well understood. The magnitude of the analog audio signal is converted into a digital representation, and the digital audio sample includes a number of bits that conveys that digital representation. In certain embodiments, one digital audio sample is represented by sixteen bits (although the number of bits in practice may be more or less than sixteen if so desired). In accordance with some embodiments, the sampling rate for digital audio samples is within the range of about 16,000 to 48,000 samples per second, and certain embodiments utilize a sampling rate of 32,000 samples per second.

As used here, a "block" of samples refers to a set, group, or other collection of samples contained in a media stream. The number of samples in a block can be arbitrarily defined, or the number may be selected for compatibility with certain data communication standards, streaming media standards, hardware requirements, and/or software requirements. In certain embodiments, one block of digital audio samples is represented by 1024 consecutive samples (although the number of samples per block may be more or less than 1024 if so desired). Consequently, for a sampling rate of 32,000 samples per second, one block of digital audio samples represents about 32 ms of time.

The process **300** is able to respond in a real-time and dynamic manner to accommodate different media streams (the end user may switch from one media stream to another for presentation at the media player). In this regard, the process **300** begins by initializing processing of a media stream and entering a fast adaptation mode (task **302**). The fast adaptation mode represents an initial training or learning period for the gain control technique described herein. In practice, the fast adaptation mode quickly adjusts the gain of the initial digital audio samples in the media stream so that the volume perceived by the user can be normalized (if needed) by at least some amount. Thereafter, the encoder system can transition to a steady state mode during which the gain control is adjusted in a more accurate and controlled manner.

For processing of digital audio samples during the fast adaptation mode, the encoder system utilizes a designated set of weighting factors; these weighting factors are used to calculate the gain to be applied to blocks of digital audio samples. The weighting factors control the extent to which the gain applied to adjacent blocks of digital audio samples can differ. The weighting factors and exemplary gain calculation methodologies are described in more detail below with reference to FIG. **4**. In practice, the weighting factors are empirically determined values that are accessible by the encoder system. Although any number of weighting factors could be used, this particular embodiment uses two weighting factors, which are labeled w_1 and w_2 herein. Moreover, the values of w_1 and w_2 are variable for the fast gain adaptation scheme, and the values of w_1 and w_2 are fixed for the steady state gain adaptation scheme. For the embodiment described here, w_1 and w_2 are positive and satisfy the relationship $w_1+w_2=1.0$ in both fast adaptation and steady state modes of operation.

In certain situations, the value of w_1 for the fast adaptation mode is less than or equal to the value of w_1 for the steady state mode, and the value of w_2 for the fast adaptation mode is greater than or equal to the value of w_2 for the steady state mode. In certain situations, $w_1>w_2$ for both the fast adaptation mode and the steady state mode. In certain situations, the difference w_1-w_2 for the fast adaptation mode is less than the difference w_1-w_2 for the steady state mode. As one specific non-limiting example, in the fast adaptation mode, $w_1=0.7$ and $w_2=0.3$, and in the steady state mode, $w_1=0.9$ and $w_2=0.1$.

For certain embodiments, the value of w_1 in the fast adaptation mode is always less than the value of w_1 in the steady state mode (although at the transition between modes the value might be the same), the value of w_2 in the fast adaptation mode is always greater than the value of w_2 in the steady state mode (although at the transition between modes the value might be the same), and the value of w_1 is always greater than the value of w_2 (for both modes). Moreover, the values of w_1 and w_2 need not remain constant during the fast adaptation mode. Indeed, these values can be adjusted during the fast adaptation mode to arrive at the values to be used during the steady state mode. In a typical implementation, w_1 increases linearly from one value (for example, 0.7) to another value (for example, 0.9), and w_2 decreases linearly from one value (for example, 0.3) to another value (for example, 0.1) during the fast adaptation mode.

For example, suppose that the process **300** remains in the fast adaptation mode for a hundred blocks of audio samples. When the process **300** enters the fast adaptation mode, the values of w_1 and w_2 are initialized to 0.7 and 0.3, respectively. After the gain adjustment of the first block of audio samples, w_1 is increased by

$$\frac{0.9-.07}{100}$$

and w_2 is decreased by

$$\frac{0.3-0.1}{100}.$$

This modification of w_1 and w_2 happens after gain adjustment of each block of digital audio samples as long as the process **300** remains in the fast adaptation mode. By the time the process **300** reaches the end of the fast adaptation mode, the linear adjustments of w_1 and w_2 result in values of 0.9 and 0.1, respectively. Thereafter, the process **300** enters the steady state mode with these values. In the steady state adaptation mode, w_1 and w_2 do not undergo any further changes and they remain constant at their respective values (0.9 and 0.1 for this example). Later at some point in time, when the system switches back to the fast adaptation mode, w_1 and w_2 are again reset to their initial values (0.7 and 0.3 for this example) and linear adjustment occurs as explained above.

Referring back to FIG. **3**, the process **300** can compute or access the gain weighting factors for the fast adaptation mode (task **304**), and obtain the next block of digital audio samples for processing (task **306**) during the fast adaptation mode. The process **300** then continues by calculating a gain value for the current block and adjusting the gain of the digital audio samples in the current block in accordance with the calculated gain value (task **308**). Such adjustment or modification of the digital audio samples results in gain-adjusted digital audio samples having normalized volume (loudness) during presentation at the destination media player. During task **308**, the gain value for the current block is calculated using the currently applicable weighting factors (w_1 and w_2) corresponding to the fast adaptation mode. In this regard, the gain of the digital audio samples is adjusted using a fast gain adaptation scheme for as long as the process **300** remains in the fast adaptation mode. For the embodiments described here, the gain value represents a multiplicative gain that is used as a multiplier for the original non-adjusted digital audio sample value. Thus, a gain value of one represents no change and the original digital audio sample value will remain unchanged with a gain value of one.

The process **300** transmits the gain-adjusted digital audio samples to the remotely-located digital media player (task **310**) in an ongoing manner. In certain embodiments, task **310** transmits the gain-adjusted digital audio samples in blocks, as is well understood. Moreover, the gain-adjusted digital audio samples will typically be transmitted in a media stream that also includes or otherwise conveys video content. Upon receipt, the media player simply decodes and presents the media stream to the user as usual. The media player need not perform any additional or special processing to implement the volume normalizing technique described here because the digital audio samples received by the media player are already gain-adjusted.

As mentioned previously, the fast adaptation mode is utilized as a brief training or learning period for new media streams. Accordingly, the encoder system may determine, detect, or otherwise be instructed to switch from the fast adaptation mode to the steady state mode (query task **312**). If the process **300** detects a mode switching condition, then it can enter the steady state mode (task **314**). Otherwise, the process **300** can return to task **304** to compute or access the newly adjusted values of w_1 and w_2, obtain the next block of audio samples for processing, and continue as described previously. The mode switching condition can be associated with one or any number of appropriate metrics, measures, or parameters. For example, the fast adaptation mode may remain active for a predetermined time period after initializing the processing of the current media stream, for a predetermined time period after the system enters the fast adaptation mode, for a predetermined time period after the weighting factors are computed or retrieved in task **304**, etc. In typical implementations, the fast adaptation mode lasts for about four to eight seconds. As another example, the fast adaptation mode may remain active for a predetermined number of blocks (or samples) after initializing the processing of the current media stream, for a predetermined number of blocks (or samples) after the system enters the fast adaptation mode, for a predetermined number of blocks (or samples) after the weighting factors are retrieved in task **304**, etc.

Assuming that it is time for the encoder system to switch modes, the process will enter and initiate the steady state mode. The steady state mode represents a "long term" and relatively stable period for the gain control technique described herein. In practice, the steady state mode takes over after the fast adaptation mode has made its initial gain adjustments. During the steady state mode, the gain of the digital audio samples is adjusted in an ongoing and accurate manner so that the volume perceived by the user remains normalized (if needed) relative to the reference volume level.

For the steady state mode, the process **300** retrieves or accesses the gain weighting factors for the steady state mode (task **316**), which are different than the gain weighting factors used during the fast adaptation mode. The process **300** also obtains the next block of digital audio samples for processing (task **318**) during the steady state mode. The process **300** then continues by calculating the gain value for the current block and adjusting the gain of the digital audio samples in the current block in accordance with the calculated gain value (task **320**). During task **320**, the gain value for the current block is calculated using the gain weighting factors (w_1 and w_2) corresponding to the steady state mode. Therefore, the gain of the digital audio samples is adjusted using a steady state gain adaptation scheme for as long as the process **300** remains in the steady state mode, where the steady state gain adaptation scheme is different than the fast gain adaptation scheme. The process **300** transmits the gain-adjusted digital audio samples to the remotely-located digital media player (task **322**) as described above for task **310**. Again, the media player need not perform any additional or special processing to implement the volume normalizing technique described here because the digital audio samples received by the media player are already gain-adjusted.

As mentioned previously, the process **300** can be repeated for each new media stream. Thus, the encoder system may determine, detect, or otherwise be instructed to switch from the current audio stream to a new audio stream (query task **324**). If the process **300** detects a new media or audio stream, then it can initialize the processing of the new media stream and again enter the fast adaptation mode (task **302**). Otherwise, the process **300** can return to task **318**, obtain the next block of audio samples for processing in the steady state mode, and continue as described previously.

Although the embodiment described here uses two modes (fast adaptation and steady state), an adaptive digital gain control technique could instead employ only one mode, or it could employ more than two different modes. The use of two different modes strikes a good balance between audio quality, effectiveness, and normalization speed.

Referring now to FIG. **4**, the digital gain calculation process **400** can be utilized by the encoder system during the adaptive digital gain control process **300**. The process **400** is performed during both the fast adaptation mode and the steady state mode (with different values for the weighting factors w_1 and w_2, as explained previously). The process **400** may begin (task **402**) with the first block of digital audio samples (where k indicates the block number), and by obtaining that block of digital audio samples in the media stream for processing (task **404**). The process **400** considers the digital audio samples in blocks because a single audio sample conveys no inherent loudness or volume information by itself. For this embodiment, the process **400** calculates a loudness estimate (L_k) for the current block of digital audio samples (task **406**), where the kth block includes N samples: $\{a_1, a_2, a_3, \ldots a_N\}$. Although other estimating methodologies could be employed, the embodiment described here calculates the loudness estimate in accordance with the expression

$$L_k = \sum_{i=1}^{i=N} |a_i|,$$

where L_k is the loudness estimate, a_i represents the digital audio samples, and the current block includes N digital audio samples. The absolute value of each audio sample is taken because any given audio sample may be positive or negative, depending upon its intended sound pressure direction relative to the listener's eardrums. Thus, the process **400** calculates the loudness estimate as a sum of the "magnitudes" of the audio samples contained in the current block.

The calculated loudness estimate can then be compared to a silence threshold value (query task **408**). The silence threshold value may be empirically determined and defined such that it is low enough to serve as an accurate threshold and high enough to contemplate bit errors, artifacts, inconsistencies in the original audio data, and "non-zero" audio samples that cannot be detected as sound. If the calculated loudness estimate is less than the silence threshold, then the process **400** can apply a multiplicative gain of one (or any baseline value, which may but need not be approximately equal to one) to the current block of digital audio samples (task **410**). In other words, the process **400** assumes that the gain value (g_k) for the current block will be equal to one. Thus, if the process **400** determines that the current block represents silence, then there is no need to apply any gain, and the remainder of the process **400** can be bypassed. As explained above with reference to FIG. **3** and the process **300**, while operating in the fast adaptation mode, the values of w_1 and w_2 are linearly updated on a block-by-block basis (task **412**). In this regard, task **412** leads back to task **404** so that the process **400** can obtain the next block for processing. If in the steady state mode, then task **412** would be bypassed.

If query task **408** determines that the loudness estimate is not less than the silence threshold, then the process **400** may continue by determining or calculating a reference gain value (g_r) that is influenced by the loudness estimate (task **414**). More specifically, the reference gain value is based upon the loudness estimate and a reference loudness value. Although other methodologies could be employed, the embodiment described here calculates the reference gain value in accordance with the expression

$$g_r = \frac{L_{ref}}{L_k}.$$

where g_r is the reference gain value, and L_{ref} is the reference loudness value. The reference loudness value is a constant value that represents, corresponds to, or otherwise indicates the desired normalized volume. Ideally and theoretically, therefore, gain-adjusted digital audio samples will be characterized by an adjusted loudness that corresponds to the reference loudness value. The reference gain value is used later in the process **400**.

The process **400** may also calculate a maximum gain value (g_m) for the current block of digital audio samples (task **416**). Although other methodologies could be employed, the embodiment described here calculates the maximum gain value in accordance with the expression

$$g_m = \frac{2^{n-1}}{a_{max}},$$

where n is the number of bits per digital audio sample, and where a_{max} is the maximum absolute sample value in the current block of digital audio samples. The process **400** determines the maximum allowable gain value for the current block in this manner to prevent bit overflow in the digital audio samples of the block. For example, if the current block includes a digital audio sample that has a relatively high value that approaches the maximum sample value, then very little multiplicative gain can be applied to that sample without causing overflow. On the other hand, if all of the samples in the block have relatively low values, then a higher amount of multiplicative gain can be applied to the block.

This particular embodiment also determines or calculates an estimated gain value (g_e) for the current block of digital audio samples (task **418**), where the estimated gain value is influenced by the reference gain value and/or by the maximum gain value. More specifically, the estimated gain value is based upon the reference gain value and the maximum gain value. Although other techniques could be employed, the embodiment described here calculates the estimated gain value in accordance with the expression $g_e=\min(g_r, g_m)$, where g_e is the estimated gain value. Thus, the estimated gain value will be equal to either the reference gain value or the maximum gain value, whichever one is lower (or equal to both if they are the same).

Next, the process **400** calculates the gain value (g_k) to be applied to the current block of digital audio samples (task **420**). Although other methodologies could be employed, the embodiment described here calculates the estimated gain value in accordance in accordance with the expression $g_k=\min(w_1 \times g_{k-1}+w_2 \times g_e, g_m)$, where g_k is the computed gain value, and w_1 and w_2 are the weighting factors, which were described previously. The expression for g_k includes a minimum operator that selects one of two values, whichever is lower (or selects either value if they are both the same). The first value is defined by the term $w_1 \times g_{k-1}+w_2 \times g_e$, and the second value is the maximum gain value (g_m). As indicated by this expression, the gain value will be influenced by the estimated gain value, the maximum gain value, and a previous gain value (g_{k-1}) for a previous block of digital audio samples in the media stream. For the reasons described above, this computed gain value will also be influenced by the reference gain value calculated during task **414**. For this particular embodiment, the gain value for the current block (g_k) is determined in response to the gain value for the immediately preceding block (g_{k-1}). Alternatively (or additionally), g_k could be calculated by considering other gain values for blocks prior to the immediately preceding block. This reliance on a previous gain value prevents large block-to-block variations in the applied gain. In practice, g_k will have a value that ranges from 1.0 to about 8.0, although values that exceed 8.0 might be realized in certain embodiments.

The computed gain value for the current block of digital audio samples can then be applied to the samples in the block (task **422**). In practice, task **422** modifies, adjusts, or otherwise changes the current block of digital audio samples. More specifically, task **422** modifies each sample in the current block by multiplying the original sample value by g_k, resulting in a gain-adjusted sample value. As mentioned above with reference to the process **300**, the gain-adjusted sample values are sent to the remotely-located media player, which can then present the media stream with a normalized loudness. FIG. **4** depicts task **422** leading back to task **412** as an indication of the potentially ongoing block-by-block nature of the process **400**. As described previously, the process **400** can be initially performed for the fast adaptation mode (using the linearly adjusted set of weighting factors) and then repeated for the steady state mode (using a fixed set of weighting factors).

While at least one exemplary embodiment has been presented in the foregoing detailed description, it should be appreciated that a vast number of variations exist. It should also be appreciated that the exemplary embodiment or embodiments described herein are not intended to limit the scope, applicability, or configuration of the claimed subject matter in any way. Rather, the foregoing detailed description will provide those skilled in the art with a convenient road map for implementing the described embodiment or embodiments. It should be understood that various changes can be made in the function and arrangement of elements without departing from the scope defined by the claims, which includes known equivalents and foreseeable equivalents at the time of filing this patent application.

What is claimed is:

1. A non-transitory and tangible computer-readable medium having instructions operable to cause a digital media processing device to perform operations for a media stream, comprising:

calculating, by the digital media processing device, a loudness estimate for a current block of digital audio samples in the media stream, the loudness estimate calculated as a sum of magnitudes of the digital audio samples contained in the current block;

calculating, by the digital media processing device, a reference gain value for the current block of digital audio samples, the reference gain value being equal to a constant reference loudness value divided by the loudness estimate;

calculating, by the digital media processing device, a maximum gain value for the current block of digital audio samples, based on a maximum absolute sample value in the current block of digital audio samples;

choosing, by the digital media processing device, an estimated gain value for the current block of digital audio samples, the estimated gain value being equal to either the reference gain value or the maximum gain value, whichever is less;

selecting, by the digital media processing device, either a first current gain value or a second current gain value for the current block of digital audio samples, whichever is less, the first current gain value comprising a weighted sum of the estimated gain value and a previous gain value applied to a previous block of digital audio samples in the media stream, and the second current gain value comprising the maximum gain value, to obtain a selected current gain value;

modifying, by the digital media processing device, the current block of digital audio samples by applying the selected current gain value to the digital audio samples in the current block of digital audio samples, resulting in gain-adjusted digital audio samples; and

transmitting, by the digital media processing device, the gain-adjusted digital audio samples to a remotely-located media player to generate sound based on the gain-adjusted digital audio samples.

2. The computer-readable medium of claim **1**, wherein the computer program product is operable to cause the digital media processing device to perform further operations, comprising:

comparing the loudness estimate to a silence threshold value; and

applying a multiplicative gain value of one to the current block of digital audio samples when the loudness estimate is less than the silence threshold value.

3. The computer-readable medium of claim **1**, wherein calculating the loudness estimate is performed in accordance with the expression

$$L_k = \sum_{i=1}^{i=N} |a_i|,$$

where L_k is the loudness estimate, a_i represents the digital audio samples, and the current block includes N digital audio samples.

4. The computer-readable medium of claim **3**, wherein calculating the reference gain value is performed in accordance with the expression

$$g_r = \frac{L_{ref}}{L_k},$$

where g_r is the reference gain value, and L_{ref} is the constant reference loudness value.

5. The computer-readable medium of claim **4**, wherein:

calculating the maximum gain value is performed in accordance with the expression

$$g_m = \frac{2^{n-1}}{a_{max}},$$

where g_m is the maximum gain value, n is the number of bits per digital audio sample, and a_{max} is the maximum absolute sample value in the current block of digital audio samples; and

choosing the estimated gain value is performed in accordance with the expression $g_e=\min(g_r, g_m)$, where g_e is the estimated gain value.

6. The computer-readable medium of claim **5**, wherein the selecting is performed in accordance with the expression $g_k=\min(w_1 g_{k-1}+w_2 g_e, g_m)$, where g_k is the selected current gain value, and w_1 and w_2 are weighting factors.

7. The computer-readable medium of claim **1**, wherein modifying the current block of digital audio samples is performed for a predetermined time period after initializing processing of the media stream.

8. The computer-readable medium of claim **1**, wherein modifying the current block of digital audio samples is performed for a predetermined number of blocks of the digital audio samples after initializing processing of the media stream.

9. The computer-readable medium of claim **1**, wherein:

one digital audio sample is represented by sixteen bits; and

one block of digital audio samples is represented by 1024 digital audio samples.

10. A method of performing operations on a media stream, the method comprising:

calculating, by a digital media processing device, a loudness estimate for a current block of digital audio samples in the media stream, the loudness estimate calculated as a sum of magnitudes of the digital audio samples contained in the current block;

calculating, by the digital media processing device, a reference gain value for the current block of digital audio samples, the reference gain value being equal to a constant reference loudness value divided by the loudness estimate;

calculating, by the digital media processing device, a maximum gain value for the current block of digital audio samples, based on a maximum absolute sample value in the current block of digital audio samples;

choosing, by the digital media processing device, an estimated gain value for the current block of digital audio samples, the estimated gain value being equal to either the reference gain value or the maximum gain value, whichever is less;

selecting, by the digital media processing device, either a first current gain value or a second current gain value for the current block of digital audio samples, whichever is less, the first current gain value comprising a weighted sum of the estimated gain value and a previous gain value applied to a previous block of digital audio samples in the media stream, and the second current gain value comprising the maximum gain value, to obtain a selected current gain value;

modifying, by the digital media processing device, the current block of digital audio samples by applying the selected current gain value to the digital audio samples in the current block of digital audio samples, resulting in gain-adjusted digital audio samples; and

transmitting, by the digital media processing device, the gain-adjusted digital audio samples to a remotely-located media player to generate sound based on the gain-adjusted digital audio samples.

11. The method of claim **10**, further comprising:

comparing the loudness estimate to a silence threshold value; and

applying a multiplicative gain value of one to the current block of digital audio samples when the loudness estimate is less than the silence threshold value.

12. The method of claim **10**, wherein calculating the loudness estimate is performed in accordance with the expression

$$L_k = \sum_{i=1}^{i=N} |a_i|,$$

where L_k is the loudness estimate, a_i represents the digital audio samples, and the current block includes N digital audio samples.

13. The method of claim **12**, wherein calculating the reference gain value is performed in accordance with the expression

$$g_r = \frac{L_{ref}}{L_k},$$

where g_r is the reference gain value, and L_{ref} is the constant reference loudness value.

14. The method of claim **13**, wherein:

calculating the maximum gain value is performed in accordance with the expression

$$g_m = \frac{2^{n-1}}{a_{max}},$$

where g_m is the maximum gain value, n is the number of bits per digital audio sample, and a_{max} is the maximum absolute sample value in the current block of digital audio samples; and

choosing the estimated gain value is performed in accordance with the expression $g_e = \min(g_r, g_m)$, where g_e is the estimated gain value.

15. The method of claim **14**, wherein the selecting is performed in accordance with the expression $g_k = \min(w_1 g_{k-1} + w_2 g_e, g_m)$, where g_k is the selected current gain value, and w_1 and w_2 are weighting factors.

16. The method of claim **10**, wherein modifying the current block of digital audio samples is performed for a predetermined time period after initializing processing of the media stream.

17. The method of claim **10**, wherein modifying the current block of digital audio samples is performed for a predetermined number of blocks of the digital audio samples after initializing processing of the media stream.

18. The method of claim **10**, wherein:

one digital audio sample is represented by sixteen bits; and

one block of digital audio samples is represented by 1024 digital audio samples.

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